REMARKS

This responds to the Office Action mailed on August 12, 2005.

Claims 1, 12, 17 and 23 are amended, no claims are canceled or added; as a result, claims 1-33 remain pending in this application.

Interview Summary

Applicant thanks Examiner Andrew R. Graham for the courtesy of a telephone interview on October 4, 2005 with Applicant and Applicant's representative Garth Vivier. The invention in general was discussed and argument was presented regarding the novel features of claim 1. However, no agreement was reached.

During the Examiner Interview the arguments raised in the Office Action concerning the use of the word "capable" was discussed. It will be noted that claim 1 has been amended to read "controllably converting said input sample rate to any one of said plurality of differing intended output sample rates for any output data sample in response to a variation in an intended rational sample rate conversion ratio by converting said input sample rate associated with said input signal to any one of the plurality of differing intended output sample rates...." Accordingly, the word "capable" has been removed (without prejudice) and the wording at the end of the second limitation of claim 1 has been added to the beginning of the limitation. It is submitted that the aforementioned restatement renders the above remarks in the Office Action moot.

It was submitted by the Applicant that jitter controlled by means of a Phase Lock Loop (PLL) does not in any way anticipate the invention as claimed. First, the sample rate in claim 1 is varied by **interpolation** and not by a PLL and, accordingly, claim 1 cannot read on to jitter compensation by use of a PLL as described in Pohlmann. Second, it will be noted that the conversion by interpolation is to a plurality of **intended** output sample rates and not random samples rates arising from jitter. To further clarify the invention, claim 1 has been amended to include the limitation of an intended **rational** sample rate conversion ratio.

The Applicant also discussed the significance of the distributed or multistage filtering as claimed in claim 1. Claim 1 includes the limitations of "**a first transition band** having an image corresponding thereto" and converting the sample rate "by interpolation with an interpolator

having associated therewith **a second transition band, with the width associated with said second transition band being a function of a spectral separation of said first transition band and said image.**" It is submitted that this limitation, as claimed in the context of claim 1, is not described or even suggested by Chester (U.S. 5,717,617) or any other prior art.

§ 1.132 Declaration

A § 1.132 Declaration supporting the non-obviousness of the invention claimed in the present application accompanies this Response and Amendment to the Final Office Action mailed August 12, 2005.

Information Disclosure Statement

Applicant submitted an Information Disclosure Statement and a 1449 Form on May 28, 2003. Applicant respectfully requests that an initialed copy of the 1449 Form be returned to Applicants' Representatives to indicate that the cited references have been considered by the Examiner.

§102 Rejection of the Claims

Claims 1-10 and 12-31 were rejected under 35 U.S.C. § 102(b) for anticipation by Chester (U.S. 5,717,617).

The invention as claimed in claim 1 includes the limitations of 1) a sample rate conversion ratio which can be changed for each output sample and 2) multistage filtering with first and second transition bands.

It is submitted that the following limitations of claim 1 are neither described nor even suggested in Chester:

> "*for said plurality of data points*, controllably converting said input sample rate to any one of said plurality of differing intended output sample rates *for any output data sample* in response to a variation in an intended rational sample rate conversion ratio"; and

"by interpolation with an interpolator having associated therewith a second transition band, with the width associated with said second transition band being a function of a spectral separation of said first transition band and said image"

As discussed during the Examiner interview, it is strongly submitted that Chester does not describe or even suggest the limitation of "***for said plurality of data points***, controllably converting said input sample rate to any one of said plurality of differing output sample rates ***for any output data sample*** in response to a variation in an intended rational sample rate conversion ratio." Thus, for a given set of data points, the sample rate can be converted on a sample-by-sample basis or dynamically (on-the-fly) without reconstruction of the architecture of the sample rate converter.

In contrast, Chester requires construction of a **new architecture** (and thus not on a sample-by-sample basis) in which the coefficients for the filters, the number of stages, and the values of L and M must be readjusted. He gives no indication whatsoever that this can be done while operating on the same input sequence. This fact is made clear in a number of places in Chester's specification, such as the description and table beginning at column 4 line 46.

The Final Office Action of August 12, 2005 on page 9 submits that the limitation in claim 1 of "controllably varying" (amended to "controllably changing") is disclosed by Pohlmann 57-60. As discussed during the Examiner Interview, Pohlmann teaches adjusting jitter using a Phase Lock Loop (PLL) whereas the invention as claimed performs sample rate conversion using an "interpolator." In particular claim 1 includes the limitation "by converting .... to any one of the plurality of differing **intended** output sample rates **by interpolation**" Accordingly, it is submitted that this reference to Pohlmann and the associated jitter arguments presented in the Office Action are not applicable. Further, the jitter in Pohlmann is generally considered to be a source of noise, distortion or error, it can instead by analyzed as an instantaneous (and typically random) change in the sample rate. The invention as claimed in claim 1 is directed to the conversion input signal at an input sample rate to one of a plurality of differing **intended** output sample rates.

In view of the above it is submitted that Chester does not disclose each and every limitation of claim 1. Further, in view of the remarks above, it is submitted that Chester does not disclose each and every limitation of claim 17.

Applicant also submits that the invention, as claimed in claims 1 and 17, are not obvious in view of Chester and Pohlmann or any other prior art references.

The three known existing and independent techniques for sample rate conversion known in the art are as follows.

The first is that of computationally efficient half-band FIR and IIR filters. These filters provide for particularly computationally efficient sample rate conversion when the sample rate conversion ratio is exactly 2 or exactly ½. They **do not allow for changing of the sample rate conversion ratio for each output sample**, and their transfer function characteristics are generally optimized to a single parameter which trades off the steepness of the transition band to the amount of passband ripple and stopband rejection.

The second independent technique is that of multistage filtering. This is well known in sample rate conversion for **fixed rational conversion ratios**. It is generally used when the sample rate conversion ratio is much greater than 1 or much less than 1, although it is also useful for some ratios L/M where L/M is near 1 yet both L and M are large. **This technique is not known to be applicable when the sample rate conversion ratio can be changed for each output sample.**

The third independent technique is that of multipoint variable sample rate interpolation, as originally proposed by Smith and Gossett, and as improved by Rossum as described in "Constraint Based Audio Interpolators" presented at IEEE Workshop on Applications of Signal Processing to Audio and Acoustics and cited in a previous information disclosure statement in this application. This technique allows for sample rate conversion with a sample rate conversion ratio which can be changed for each output sample and an interpolator transfer function, which is

a function of the convolution coefficients. This is an extremely flexible technique, which allows sample rate conversion at any rational sample rate conversion ratio at arbitrary precision. **This technique is not described in any way with reference to multistage filtering.**

The prior art includes the combination of halfband filtering and multistage filtering whenever a the sample rate conversion ratio can be factored by 2 or ½. For example, a multistage sample rate converter providing a sample rate of 12 times the input might employ two halfband filters each increasing the rate by 2, and a final stage multiplying this fourfold increase by 3. However, multipoint variable sample rate interpolation is an independent techniques that is used exclusively and no suggestion is provided anywhere in the prior art to combine it with either of the first two techniques, as discussed in more detail below.

1. Existing references and prior art do not suggest combining the prior art methods to create the invention as claimed, nor provide any suggestion or motivation to combine the prior art.

In the current invention, as claimed, the example embodiments all include an interpolator, for example an appropriately designed Nth order sum of products FIR filter with linear interpolation of coefficients (see for example the specification on page 20, lines 7-19). For example, the embodiment of Figure 6 includes this element as interpolator 604.

The seminal reference for this type of interpolator is "A Flexible Sampling-Rate Conversion Method" by Julius O. Smith and Phil Gossett (Proc. IEEE Int. Conf. Acoust., Speech, Signal Processing, San Diego, CA, March 1984 and cited in an Information disclosure statement in this application). This paper states in its abstract:

> One filter table handles all conversion factors. Formulas are given for determining the size of the lookup table needed for a given precision requirement.

Thus this paper clearly indicates that the technique described is applicable to **ALL** sample rate conversion ratios, and that the technique is applicable to **ANY** given precision

requirement by adjustment of the size of the lookup table. In other words, there is no suggestion of any motivation to combine the techniques described therein with any other known signal processing techniques to obtain better precision or better performance. Instead, per the title, the technique is considered flexible enough to stand on its own.

To the Applicant's knowledge, there are no other papers or references that suggest combining the techniques of Smith or similar techniques to form the current invention as claimed, nor any that suggest there would be a motivation to make such a combination.

The Applicant believes, based on his extensive personal experience, that this fact is the primary reason that the benefits of his invention were not realized by others skilled in the art. Those skilled in the art have believed that when the flexibility in sample rate conversion offered by the technique of Smith was required (the ability to change the sample rate conversion ratio for each output sample), the optimal solution was Smith's technique and any attempt to combine it with other techniques would simply be a waste of computational power.

For at least the above reasons, Applicant submits that the invention as claimed is non-obvious.

2. Existing references teach away from the combination of the prior art methods to form the current invention as claimed.

While using a series of filters for sample rate conversion is well known in the prior art, existing references teach away from using such a multistage approach in the context of the current invention as claimed. In particular, the architecture of the invention, as claimed in claim 1, allows for an arbitrary rational sample rate conversion ratio, which can be changed for each output sample. Because the architecture of the claimed invention allows arbitrary rational sample rate conversion ratios, this includes sample rate conversion ratios that are approximately

equal to one. The architecture of claim 1 provides substantial benefit in the computational requirements for a given level of precision compared to a single stage approach.

Yet in the highly respected text "Digital Processing of Speech Signals" by Rabiner and Schafer (Prentice-Hall, 1978 with many subsequent reprints), it is stated on page 31:

> "Assuming that the required filtering is being performed using FIR filters, then for large changes in the sampling rate (i.e. large M for decimators, or large L for interpolators) it has been shown that it is more efficient to reduce (or increase) the sampling rate with a **series of decimator stages than to make the entire rate reduction with one stage**."

This clearly leads to the conclusion that a multistage approach is beneficial when the sample rate conversion ratio is much greater than or much less than one, but **NOT** when it is near unity.

It is also stated in the same text on page 30:

> "An extremely important consideration in the implementation of decimators and interpolators is the choice of the type of lowpass filter. For these systems, a significant savings in computation over alternative filter types can be obtained by using finite impulse response (FIR) filters in a standard direct form implementation. The savings in computations for FIR filters is dues to the observation that for decimators only one of each M output samples needs to be calculated, while for interpolators, L-1 out of every L samples of the input are zero valued, and therefore do not affect the computation. These facts **cannot be fully exploited using IIR filters**."

The filtering techniques taught in Smith make use of the FIR computational savings referenced above. Clearly Rabiner and Schafer **teach away** from combining IIR techniques when such computational savings are considered important. Since the advantage of the architecture of claim is improved computational efficiency for a given precision, this would further lead one skilled in the art away from having motivation to combine Smith with techniques that could potentially use IIR filters (see also claims 28 and 29).

Also, in a paper in the IEEE Transactions on Acoustics, Speech, and Signal Processing in April 1985, "A Note on the Computational Efficiency of the FIR Multistage Implementation of Fractional Sampling Rate Conversion" by N. S. Vasanthavada, it is stated:

> "It is the purpose of this correspondence to show the existence of some fractional sampling rate conversion problems where the conversion factor L/M is approximately equal to 1, but L>>1, M>>1 for which multistage FIR implementations are computationally less efficient than a single stage implementation."

Again, since the technique of Smith is a single stage technique implementing any rational factor L/M including those approximately equal to 1, this paper **teaches away** from investigating a multistage approach in this case. This statement leads one to conclude that a technique such as Smith's wherein **ANY** ratio can be used, the multistage approach would not provide benefit.

For at least the above reasons, Applicant submits that the invention as claimed is non-obvious.

3. Existing references fail to disclose adequate details to allow one skilled in the art to effectively combine the prior art elements into a useful embodiment of the invention.

While multistage fixed sample rate conversion ratio sample rate converters are well known in the prior art and, completely independently, the technique of Smith to create a multipoint FIR-based variable sample rate conversion ratio interpolator which can change its sample rate for each output sample is also known in the prior art, it is strongly submitted that it would not be obvious to one of ordinary skill in the art to combine these two ideas.

It will be noted that in order to design a multistage filter, the transfer functions of the filters must be properly matched to produce a useful composite filter transfer function. In particular, the width of the transition band of the variable sample rate conversion ratio

interpolator must tailored to the separation of the first transition band and its image belonging to the halfband FIR or IIR filter.

While this might appear straightforward for a fixed sample rate conversion ratio multistage filter, it is most certainly not straightforward when a variable sample rate conversion ratio interpolator is involved. This is due to the fact that it is not obvious whether the filter transfer function of the variable sample rate conversion ratio interpolator tracks its input or output sample rate. In other words, if the input sample rate of a variable sample rate conversion ratio converter is Si and its output rate is So = L/M*Si, and the associated filter transition band begins at a frequency F, does F vary as L/M varies if Si remains fixed?

Consider this: In the Applicant's specification, Figure 7B shows the filter transfer function of the halfband upsampling filter 602. This transfer function is clearly fixed with respect to its relationship to the input sampling rate of signal Xn and signal Wk since Wk's rate is always twice Xn's rate. However, is the transfer function 7D of interpolator 604 fixed or variable with respect to the rate of Wk? Or fixed or variable with respect to the rate of Ym? Since the ratio of the rate of Ym to that of Wk can be varied at every output sample, it is clear from the Applicant's disclosure that the transfer function can be fixed with respect to only one of these rates, but which one? Applicant submits the answer to this question is not obvious to one of ordinary skill in the art, yet is absolutely required in order to conceive of the architecture of the claimed invention and design the required filters in such a manner as to be useful.

In actual fact, it turns out that transfer function 7D is fixed with respect to Wk, as shown in Figure 7E by the elimination of image 722 for all values of the sample rate conversion ratio.

It should be noted, however, that the transfer functions 14E and 16G do in fact vary with the sample rate conversion ratio, which while evident from the associated figures of Applicant's description, would not be obvious to one of average skill in the art.

Applicant submits that in order to be motivated to combine the known prior art to produce the invention, as claimed, and be able to see that it actually provides computational benefit for a given level of precision, one skilled in the art would need to be provided with significant guidance as to whether the transfer function of an interpolator using Smith's technique would track with the input or output sample rate. Thus, even if there were otherwise some motivation to combine Smith's techniques with those of a multistage sample rate converter, which is strongly denied, it would **not** be obvious to one of average skill in the art that such combination could be usefully accomplished.

For at least the above reasons, Applicant submits that the invention as claimed is non-obvious.

Claim 12,as amended, reads as follows:

A method for converting a digital audio signal at an initial sample rate to a different intended sample rate, the method comprising:

receiving a plurality of data points, associated with an audio signal, at [[an]] the initial sample rate;

halfband filtering said plurality of data points with a halfband filter to provide intermediate data points, said halfband filter having a first transition band with an image corresponding thereto; and

interpolating the intermediate data points with an interpolator having independently programmable parameters and a second transition band, with a width associated with the second transition band being a function of the spectral separation of said first transition band and said image, and wherein the different sample rate is provided by interpolating at least a subset of the intermediate data points based on the independently programmable parameters, and wherein said different sample rate is controllably variable at any output data sample by interpolation.

In view of the remarks above it is also submitted that claim 12 is allowable. Further, in view of the remarks above, claim 23 is also allowable.

§103 Rejection of the Claims

Claims 11, 22, 32, and 33 were rejected under 35 U.S.C. § 103(a) as being unpatentable over Chester as applied above, in view of White (U.S. 5,808,924).

Claims 11, 22, 32 and 33 are dependent (either directly or indirectly) upon claims 1,17, and 12 which are allowable. Accordingly, dependent claims 11, 22, 32 and 33 are also allowable.

CONCLUSION

Applicant respectfully submits that the claims are in condition for allowance and notification to that effect is earnestly requested. The Examiner is invited to telephone Applicant's attorney 408-705-2698 to facilitate prosecution of this application.

If necessary, please charge any additional fees or credit overpayment to Deposit Account No. 19-0743.

Respectfully submitted,

DAVID ROSSUM

By his Representatives,

SCHWEGMAN, LUNDBERG, WOESSNER & KLUTH, P.A.
P.O. Box 2938
Minneapolis, MN 55402
408-705-2698

Date 11/04/2005 By [signature]
Garth Vivier
Reg. No. 57,313

CERTIFICATE UNDER 37 CFR 1.8: The undersigned hereby certifies that this correspondence is being deposited with the United States Postal Service with sufficient postage as first class mail, in an envelope addressed to: Mail Stop RCE, Commissioner of Patents, P.O. Box 1450, Alexandria, VA 22313-1450, on this 7 day of ~~October~~ November kL 2005.

KACIA LEE Kacia Lee

Name Signature